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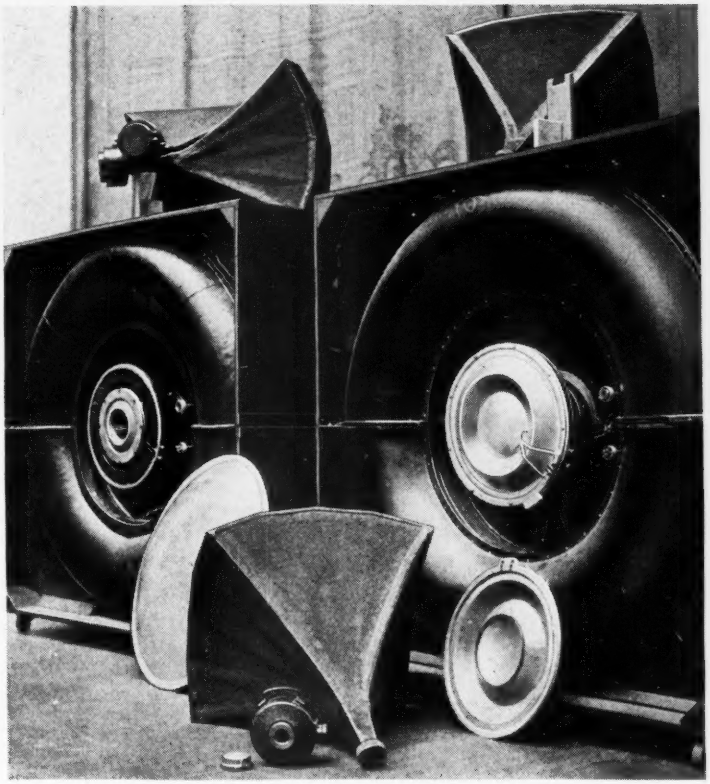

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Auditory Perspective

By W. B. SNOW
Acoustical Research

THE successful attainment* of the reproduction of music in auditory perspective is the outcome of studies carried on by the Laboratories over a number of years. The word perspective is, of course, taken over from a visual phenomenon, and implies the recognition of relative locations in space. Auditory perspective thus means the ability to judge the location and distance of sounds by the ears. It might seem that the reproduction of music would not be much improved by this ability, but it has been found by many actual tests that the majority of listeners recognize an appreciably enhanced value of the aesthetic appeal if auditory perspective is present.

One method of securing the sense of localization for reproduced sounds is to pick up the source with two microphones located in the same relative positions to each other as are a person's ears. Then by providing a separate circuit for each microphone to two head receivers held to the ears of a listener in a distant location, the directional and distance sense is completely secured. This "binaural" reproduction has already been described in the Record.** With this type of reproduction the listener is to all intents and purposes transported to the position of the pick-up microphones, and hears exactly what he would hear if he were stationed at that place. The effects that can be produced in this manner are startlingly realistic, as visitors to the Bell System exhibit at the Century of Progress Exposition will attest.

A binaural scheme for the reproduction of music before large audiences, however, would be very inconvenient. Every seat in the auditorium would have to be equipped with a pair of head phones, and in a hall of any size, the necessary wiring and its upkeep would be high. It seemed well worth while, therefore, to experiment with other methods of securing a similar effect.

It is obvious that when one listens directly to music, such as a symphony orchestra, one hears sound that—at least originally—passes through the opening between stage and auditorium. If this space were filled by an array of microphones, therefore, each of which was electrically connected by an individual circuit to a loud speaker similarly placed before an audience at a distant point, the audience would then hear—assuming perfect transmission—exactly what they would have, had they been listening directly. Such an arrangement would, of course, be impracticable, but it is quite conceivable that a much smaller number of microphones, properly placed and connected by individual circuits to a similar set of loud speakers, might produce an effect substantially similar. An extensive series of tests was run at Bell Laboratories to determine what was possible in this direction. These experiments were performed

* Record, *May*, 1933, *p.* 254
**Record, *June*, 1933, *p.* 286

with either two or three channels since it is desirable to use as few channels as possible to produce the effect desired.

In the experimental set-up, shown in Figure 1, three microphones placed in an acoustically treated room were connected by individual amplifier channels to three loud speakers concealed behind a gauze curtain in the auditorium. At a little less than three-quarters of the distance back in the auditorium were seated a group of observers. Their average position is indicated by a cross on the diagram. Most of the observers had had no previous experience with this type of reproduction, and their only instructions were to note on a sheet of paper containing a line representing the curtain, the point from which the sounds they heard seemed to come. The positions from which the sounds actually originated are indicated by Roman numerals.

Tests were carried out in this manner for five different conditions. These were compared with each other and with a direct listening test in which the sounds originated on the stage in front of the listeners. The connections of the microphones and loud speakers for the five schemes are shown at the left of Figure 2. In the first, three microphones and three loud speakers were connected by independent circuits. In the second, only two microphones and loud speakers were employed but the two circuits were independent as before. The remaining three schemes employed various forms of coupling between loud speakers or microphones. The third arrangement used three microphones but only two loud speakers—the middle microphone dividing its output equally between the two loud speakers. The fourth arrangement was the inverse of the third; three loud speakers received the output of two microphones. The fifth scheme was a combination of the third and fourth. Although three microphones and three loud speakers were employed, the middle microphone and loud speaker were coupled to the two side channels.

The results obtained under these five conditions, as well as those for direct listening, are indicated at the right side of Figure 2. The average judgment of the position of the sound is indicated by circles identified by Arabic numerals, which may be compared with the actual position indicated by Roman numerals. All five

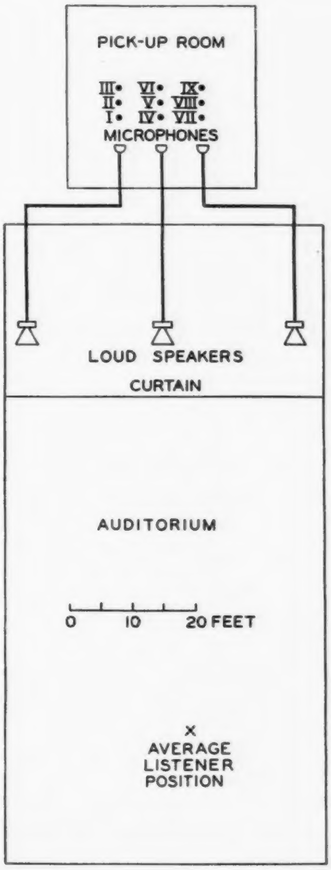


Fig. 1—For experimental purposes, pick-up microphones were mounted in a sound-proof room and connected by independent circuits to loud speakers on the stage of the auditorium

arrangements give to some extent both sidewise (angular) and depth localization, but the degree to which they correspond to the actual conditions differ. Even with direct listening, the depth localization is distinctly inaccurate.

For the three channel condition both angular and depth localization is very good although the positions at the rear of the stage seem nearer to the center than they really are. The two channel condition gave slightly wider separation for the rear positions, but on the other hand the depth localization was not so accurate. With the bridged central microphone, condition three, the apparent width of the stage remained about the same but the centered positions were brought nearer the front. A bridged center loud speaker, condition four, moved back the apparent positions of the central sounds but narrowed the apparent width of the stage. With center microphone and center loud speaker both bridged, the apparent width of stage was considerably narrowed, although the depth was somewhat improved. None of the bridged conditions were thus as good as the independent channel conditions, and three channels were appreciably better than two.

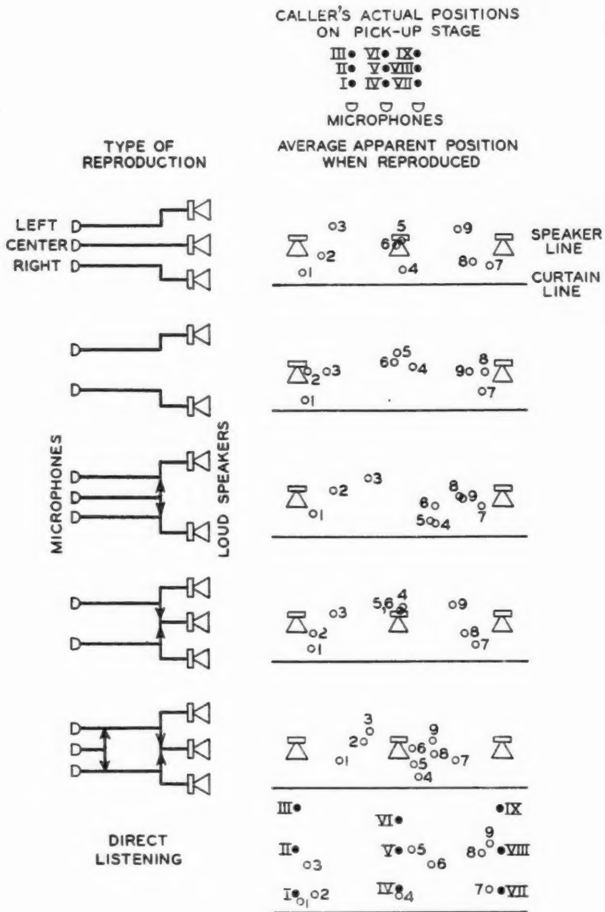


Fig. 2—Five different circuit arrangements of microphones and loud speakers were tried and the localization obtained by the listener is indicated by circles identified by Arabic numerals

The microphones on the stage receive both direct and reflected or reverberant sounds, and similarly the observers receive both direct and reverberant sound from the loud speakers. Experiment showed that decreasing either the total loudness or the amount of direct sound relative to reverberant, gave the impression that the sound was moving back on the stage. Depth localization is thus a complicated function of loudness and relative reverberation. In the two channel reproduction, for example, the center positions seemed further back because the distance of the sound from the microphones was greater, due to the lack of a central microphone. Under these conditions the ratio of direct to reverberant sound is decreased.

Angular localization on the other hand was found to depend primarily on the differ-

ence into a given angle, even though the difference is produced by a combination of similar sounds from several directions, the angle from which the reproduced sound seems to come can be obtained from the computed loudness difference by reference to Figure 3.

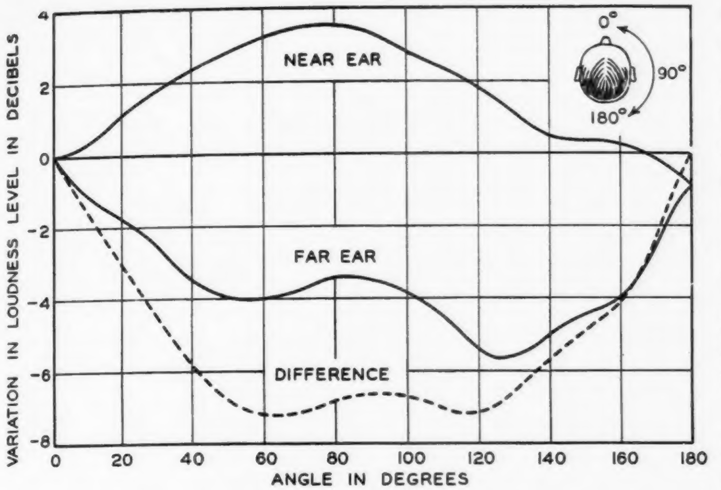


Fig. 3—Difference in loudness in the two ears for speech coming from various directions

To verify this theory for a two channel system, lines were drawn on the pick-up stage representing a constant distance ratio to the two microphones. These curves are shown on Figure 4. Since the ratio of the distances to the two microphones along any one of these curves is a contant, the difference in level at the microphones of a sound produced along on any one curve will also be constant. This difference in level, which is marked on the curve, will be carried through to the loud speakers and will cause a difference in loudness in the two ears of a listener.

These differences of loudness in the

ence in loudness of the direct sound reaching the two ears from the local speakers. Reverberation played a minor part. When one listens directly to a sound the configuration of the head causes the loudness and quality heard by each ear to differ by amounts related to the angle from which the sound comes. For speech the relationship between loudness and direction is shown by Figure 3. The ears distinguish between the front and rear angles giving identical loudness differences probably because of the quality differences. When the observer listens to the acoustic perspective system, he hears sounds from several sources all in front of him and of like quality. Calculations show that when the outputs of the loud speakers differ, resultant loudness differences are produced in the ears. If the ear always translates a given loudness differ-

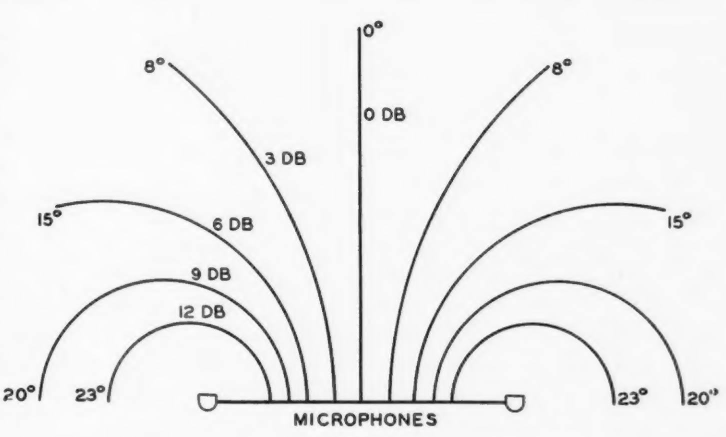


Fig. 4—Curves of constant distance ratio to the two microphones become curves of constant angular localization to the listener

two ears of a listener were calculated for a listener position along the center line of the auditorium and a distance in front of the stage equal to the separation of the loud speakers. From these calculated differences in loudness in the two ears and Figure 3, the angle from which the sound should apparently come was obtained, and these angles are also marked on the curves. A person speaking anywhere along the curve for a 6 db level difference, therefore, should appear to the listener to be at an angle of 15°. Actual tests bore out this relationship fairly closely. Moreover a speaker on the 6 db curve, appearing 15° off the center, would appear to the listener to move to an angle of 8° if the difference in level was decreased to 3 db by a manipulation of the amplifiers. In fact many of the effects of walking about the stage can be duplicated by volume manipulation as the person speaking walks back and forth along the center line of the stage. Although the observed and calculated angles agreed fairly well for central observing positions, the apparent source shifted more rapidly toward the nearer loud speaker than computation predicted, as the observer moved toward the side of the auditorium.

Similar curves were calculated and tried out for a three channel system, and a similar correlation of observed and calculated positions was found.

These curves were all calculated on the basis of sound of equal quality at the two microphones. If the quality differed materially—if one microphone picked up mostly direct sound and the other reverberant sound, for example—the localization by the observer would be quite different. It was found, for example, that if the right microphone picked up mostly direct sound and the left, reverberant sound, the sound would appear to come from the right loud speaker until the level of the left speaker was raised 10 db. In general the localization tends toward the channel giving the most natural or close-up reproduction.

These tests proved conclusively that very good localization could be obtained by a three channel system, and that two channel reproduction gave good angular localization although the distance localization was not entirely satisfactory for central positions. In the application of auditory perspective to the reproduction of orchestral music, the satisfactoriness of the two and three channel systems is even greater than indicated by the accuracy of the localization. The enhanced aesthetic appeal obtained from an auditory-perspective reproduction of an orchestra is not due so much to an accurate localization of the various sounds as to a general effect of space distribution, which adds a fullness to the overall effect. For this reason either two or three channel reproduction in auditory perspective is very satisfactory for orchestral reproductions.

Auditorium Acoustics and Control Facilities For Reproductions in Auditory Perspective

By E. H. BEDELL
Acoustical Research

WHEN music is to be reproduced in auditory perspective before a large audience, there are many requirements that must be met, and much testing and adjusting that must be done, which are not directly related to the basic problem of reproducing the complete frequency and volume ranges. One of the most important groups of adjustments is concerned with the acoustics of the halls where the music is being picked up and where it is being reproduced. The importance of the acoustic properties of an auditorium are probably not generally appreciated. Unless they are so bad that they actually spoil a reproduction or an original rendition, their existence is not usually recognized. That they play an important role under all conditions, however, could be inferred from the fact that 90% of the sound energy reaching a member of an audience may have been reflected one or more times from the various surfaces in the auditorium.

The acoustic characteristics of a hall are of particular importance when music is to be reproduced in auditory perspective, because the illusion of the actual presence of the orchestra, which it is desired to produce, depends to a large extent on the characteristics of the two halls involved. The system must be set up and adjusted to give the desired illusion under existing conditions, and in general these adjustments will differ for various auditoriums. Imperfect adjustment for

Fig. 1—For the New York demonstrations the orchestra was in a room two floors above the auditorium. The three microphones were spaced across the room in line with the conductor

the acoustics may destroy the desired illusion and be improperly ascribed to the reproducing system itself.

One of the important factors is the reverberation time of the auditoriums, and as a first step in preparation for the auditory perspective demonstration of April, 1933,* it was necessary to procure the reverberation times under various conditions of both the Academy of Music in Philadelphia, where the music was picked up, and Constitution Hall, where it was reproduced. Although in neither hall were the reverberation times for the various frequencies ideal, in both they were sufficiently satisfactory so that modifications of the halls themselves did not seem required.

In an ordinary reproduction, not in auditory perspective, one usually has the choice of reproducing the acoustic characteristics of the pick-up hall or studio—and thereby in effect transporting the listener to the pick-up location—or of allowing the acoustics of the place where the music is heard to color the reproduction, which has the effect of transporting the orchestra to the location of the listener. In an auditory perspective reproduction this choice is not possible because the objective is to give the illusion of the actual presence of the orchestra, and this requires that the acoustic coloring of the hall where the reproduction is taking place be represented, and not that of the hall where the music is picked up.

Since the room coloring is due to reflections, the pick-up microphones must therefore be placed near the orchestra if the reverberation effects of the pick-up hall are to be minimized. The position of the microphones with respect to the orchestra, as arranged for the New York demonstrations of January of this year, are shown in Figure 1. The perspective effect,

*Record *May*, 1933, *p.* 254.

obtained from the use of three channels, also requires that the microphones be placed near the orchestra. This close placement of the pick-up microphones, however, is contrary to usual practice for single channel reproduction. Here it is better to have the microphone at a greater distance from the orchestra, where more of the reverberation effects will be picked up.

When the microphones are placed close to the orchestra the effects of the acoustic characteristics of the pick-up hall are largely eliminated. The effects of the hall where the music is reproduced, however, have to be carefully studied and preserved. Studies were made in which a heterodyne oscillator, connected to the loud speakers through the amplifiers, produced tones of varying frequency which were picked up by a portable microphone connected to an automatic level recorder. The frequency was varied through the range from 35 to 15,000 cycles per second, and continuous curves of microphone response as a function of frequency were obtained for a number of positions in the auditorium. The loud speakers were placed so that each covered the entire auditorium as nearly as possible, and the curves from the automatic level recorder gave a check on these coverages. They also furnished data for the design of equalizing networks which were associated with the amplifying equipment.

In general the audience will not hear the same quality of sound that is given out by the loud speakers. This is partly due to the effects of reverberation and partly to the fact that the higher frequencies are absorbed more rapidly by the air than are the lower frequencies. The equalizers are designed, therefore, so that the best overall characteristics of the sound will, be heard at an average listener position. Correct equalization is thus different for every hall. Since the greater part of the audience in Constitution Hall was well back from the stage, equalization was based on microphone readings taken at some distance from the loudspeakers.

Besides these various provisions to insure the best quality of music and the truest illusion of the actual presence of the orchestra, tests had indicated that it was possible to produce an aesthetic effect more pleasing than that of the orchestra itself. This was accomplished by control features manipulated by a director seated in the audience. This control position, as arranged for the New York demonstrations, is shown in the photograph at the head of this article. One of the controls was a volume adjustment which permitted the output of the orchestra to be modified as the director deemed necessary, and allowed a larger range of volume from the loud speakers than was possible from the orchestra. When the orchestra was playing alone the volume of the three channels was controlled from a single dial. When a soloist was accompany-

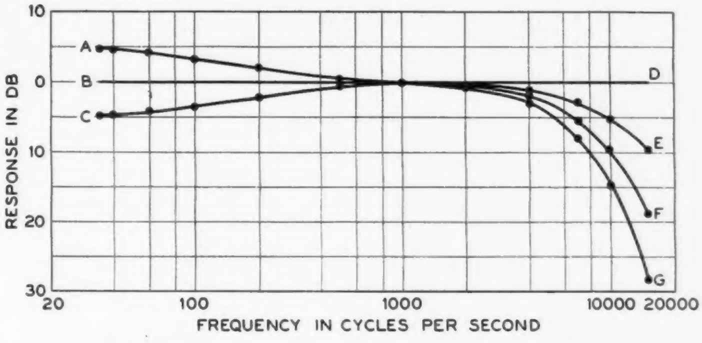


Fig. 2—Quality control networks permit the high- or low-frequency components to be modified as shown above

ing the orchestra, however, the center channel, connected to a different microphone, was used exclusively by the soloist and was controlled separately. This additional microphone, which could be switched in place of the regular microphone for the center channel, was shielded by a directional baffle so that it responded to energy received from only a rather small solid angle. In this way the voice—even during the loudest passages of music—could be kept slightly above the level of the orchestra.

In addition to this volume control, a set of quality control networks were provided, which could be switched into or out of the circuit by the keys in the lower right hand cabinet at the control position. These networks increased or decreased the higher or lower frequency components as shown by the curves of Figure 2. Similar networks were provided for all three channels. When under the control of the director, the employment of these networks may add appreciably to the aesthetic effect which the music produces on the listener.

Besides these volume and quality controls, certain auxiliary circuits were supplied to aid the smoothness of the performance. These are in a cabinet at the lower left of the control position as shown in the photograph at the head of this article. An order wire gives communication between all technical operators and the director's position, and a monitor circuit is provided in the reverse direction. The microphone for it is located before the director, and loud speakers are connected in the control rooms and on the stage with the orchestra. These enable the control operators and the orchestra to hear what goes on in the auditorium, and allows the director to speak to the orchestra. Two signal circuits are also employed. One gives the orchestra a "play" or "listen" signal, and the other is a visual signal to the assistant director leading the orchestra, which can be used during the rendition of the music.

Because of these various control features, chiefly the volume and quality controls, the reproduction of music in auditory perspective as developed by the Bell System is capable of producing musical effects that are actually superior to those obtainable from any practicable size of orchestra at the present time. The system insures that practically the complete frequency and volume range of the orchestra is reproduced, and in addition a modification and enhancement of the effect is possible under the control of the musical director. It provides facilities which permit the finest musical reproductions to be heard whenever an adequate auditorium and audience is available without the very large expense of actually bringing a first class symphony orchestra to the local auditorium.

Loud Speakers and Microphone For Auditory Perspective

By A. L. THURAS
Acoustical Research

THE reproduction of orchestral music in a large auditorium presents a number of difficult problems, particularly when a requirement is that the audience shall receive an aesthetic impression comparable to that given by a personally-present orchestra. A solution of these problems on the scale successfully attempted in Constitution Hall last April would have been impossible but for the careful laboratory measurements which had been made during previous years. These showed definitely the amount of acoustic power, the range of vibration-frequencies, and to some extent the amount and character of reverberation necessary to reproduce the music of a large orchestra without noticeable change. Accordingly, the loud speakers were designed to radiate a total of 450 watts of sound power and to respond uniformly over the range from 40 to 15,000 cycles.

For radiating frequencies as low as 40 cycles per second efficiently, a horn of large dimensions is required. This horn, in order to be more compact is preferably of the folded type, but a large folded horn transmits high frequency tones inefficiently. The loud speaker was therefore constructed in two units; one for the lower and the other for the higher frequencies. An electrical network was used to divide the current into two frequency bands, the point of division being about 300 cycles per second.

In transmitting large powers at high pressure, it is essential to consider the distortion*, in the form of higher harmonics, which may be generated in the air. At the low frequency limit,

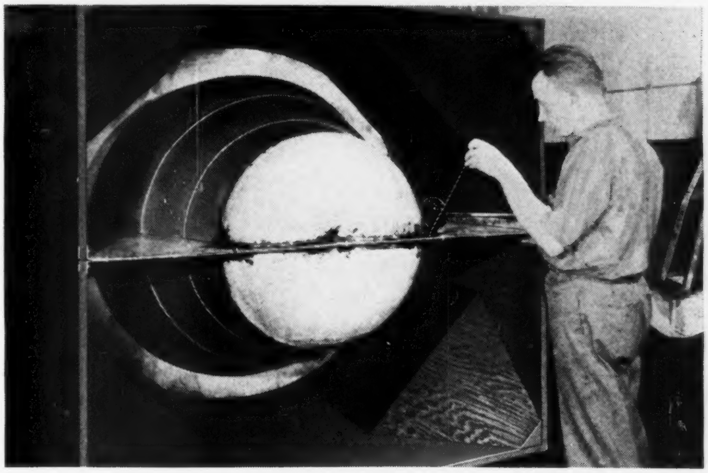


Fig. 1—To seal the joints of the low-frequency horn, melted compound is flowed around the edge by A. Trunks of the Development Shop

This distortion, pointed out by Rayleigh but neglected in his equations of wave propagation, has recently been theoretically investigated by R. Y. Rocard and applied to exponential horns. He finds that the intensity of the second harmonic increases as the square of the fundamental frequency, directly as the fundamental power, inversely as the square of the cut-off frequency of the horn, and inversely as the throat area.

each horn can radiate about three times the low frequency power of the orchestra with a second harmonic generation 30 db below the fundamental. Three low frequency units will then radiate power 9 db above that of the orchestra without noticeable distortion. At these high power levels no data are at present available on the detection of this type of distortion. Since it consists principally in adding tones an octave above the original tones, and since the ear itself generates harmonics at high pressures, it is reasonable to suppose that still higher power levels can be radiated without objectionable results.

It has been generally assumed that to avoid wide variations in output near the lower cut-off point, a horn mouth must measure across about one third the wave length of the lowest frequency. The rule here would require a mouth diameter of ten feet, but with a high efficiency receiver it is possible to use a much smaller diameter and smooth out variations by proper selection of the output impedance of the amplifier. Considering the acoustic impedance of the horn as transferred by the diaphragm to the electrical circuit, we have a generator with internal impedance—the amplifier and receiver coil—driving a variable impedance load. The problem then is to select an internal impedance such that variation of power output with load impedance shall be a minimum. This works out to be the square root of the product of the maximum and minimum values of load impedance. When this condition was met, a maximum change of horn impedance of 7.5 to one resulted in a sound output which did not vary more than 1 db.

The acoustic impedance into which a loud speaker works depends to a considerable degree on the amplitude and phase of the reflected sound waves at the horn mouth. These will in turn depend on the size of the auditorium and its acoustic damping.

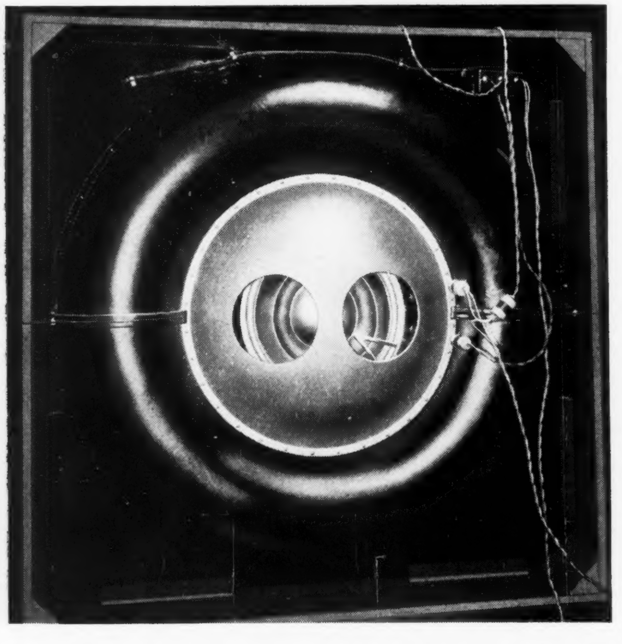


Fig. 2—Looking through holes in the cover-plate, one sees the diaphragm of the low-frequency speaker

At high frequencies the damping is great enough to attenuate the reflected sounds to a negligible amplitude. This is not the case however, in the neighborhood of forty cycles. A consideration of the phase changes with frequency in this vicinity shows that over a relatively short range of frequencies the returning waves are alternately in and out of phase with the outgoing waves. This means that the load impedance will vary according as the outgoing waves are met by aiding or opposing pressure from returning waves. Since it is still impossible to predict accurately the amounts and directions of reflected sounds, there is no way of predetermin-

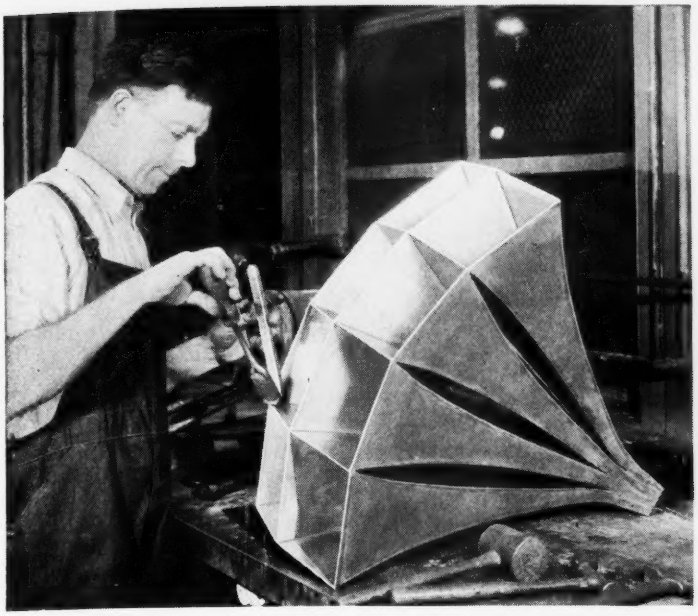


Fig. 3—W. Wynn of the Development Shop is seen here soldering one of the high-frequency horns. The sixteen horns all lead from one throat

consideration. The magnetizing coil, however, was made of copper ribbon instead of round wire which gave a considerably better space factor thus reducing the size and weight of the unit. One realizes that this is an important consideration after assembling one or two of these loud speakers.

This receiver unit and horn when connected to an amplifier in the specified manner are capable of delivering three or four times the power of an orchestra in the frequency region between 40 and 400 cycles per second; with an efficiency of about 70 per cent and a variation in sound output for a given input power to the amplifier of not more than 1 db throughout this range.

ing at just what frequencies the horn will be working into high or low impedances. Hence rather large variations in sound output would have to be tolerated were it not for the smoothing effect of properly chosen electrical impedance, as outlined in the preceding paragraph. An essential condition for this is the high efficiency of the low-frequency loud speaker.

The receiver unit was similar in construction to the Western Electric No. 555 Receiver* but much larger in size. The driving coil was made of copper ribbon, as weight in the moving element was not an important

*Record, *March*, 1928, *p.* 205

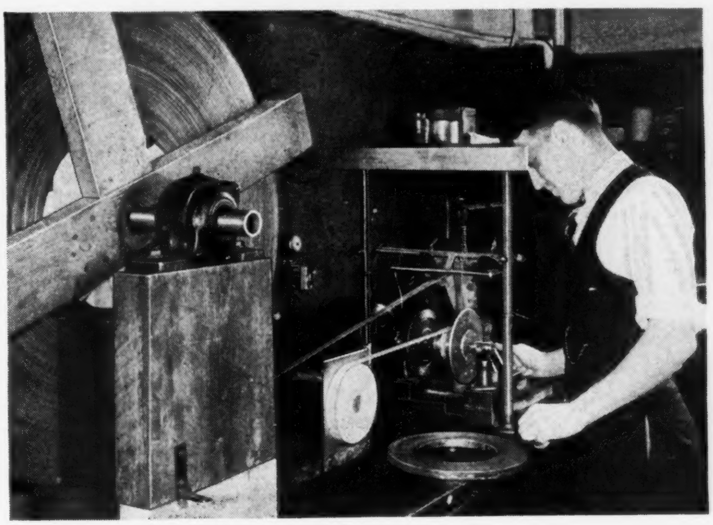


Fig. 4—Field coils of the experimental horns were wound of copper ribbon by J. H. Kronmeyer of the Acoustical Research group

The ideal distribution of sound from a horn is one in which motion of the air particles is the same in amplitude and phase throughout the solid angle subtended by the audience and so much of the walls and ceiling as are necessary for suitable reverberation. This is not realized by a tapered horn

Fig. 5—Adjustment of the acoustical resistance of the microphones was made on the basis of air-flow measurements by T. O. Osmer of Acoustical Research

of the ordinary type, which is inherently highly directive at the higher frequencies. Juxtaposition of a number of horns pointed in suitable directions can produce at their mouths a spherical wave front of dimensions large in comparison with the shortest wave length of interest, and in which the particle motions are substantially alike in amplitude and phase. To this end, a horn was designed with separate channels whose acoustic axes radiate from a common center.

Each channel has substantially an exponential taper. Toward the narrow ends these channels are brought together with their axes parallel, and are terminated in a single tapered tube which at its other end connects to the receiver unit. Sound from the latter is transmitted along the single tube as a plane wave and is divided equally among the several channels. Since the channels have the same taper, the speed of propagation of sound in them is the same. The large ends are so proportioned and placed that the particle motion of the air will be in phase and equal over the mouth of the horn. This design gives a true spherical wave front at the mouth of the horn at all frequencies for which the transverse dimensions of the mouth opening are a large fraction of a wave length.

As the frequency is increased, the ratio of wave length to transverse width of the channels becomes less and the sound will be confined more and more to the immediate neighborhood of the axis of each channel. The sound will then not be uniformly distributed over the mouth opening of the horn, but each channel will act as an independent horn. In order to have a true spherical wave front up to the highest frequencies the horn should be divided into a sufficient number of channels to make the transverse dimension of each channel small compared with the wave length up to the highest frequencies. If we wish to transmit up to 15,000 cycles it is not very practical to subdivide the horn to that extent. Both the cost of construction and the losses in the horn would be high if designed to transmit also frequencies as low as 200 cycles, as is the case under consideration, but it is not important that at very high frequencies a spherical wave front be established over the whole mouth of the horn. For this

frequency region it is perfectly satisfactory to have each channel act as an independent horn, provided that the construction of the horn is such that the direction of the sound waves coming from the channels is normal to the spherical wave front.

The angle through which sound is projected by this horn is about 60° both in the vertical and the horizontal directions. For reproducing the orchestra two of these horns, each with a receiving unit, were used. They were arranged so that a horizontal angle of 120° and a vertical angle of 60° were covered. These angular extensions were sufficient to cover most of the seats in the hall with the loud speaker on the stage. The vertical angle determines to a large extent the ratio of the direct to the indirect sound transmitted to the audience. The vertical angle of 60° was chosen purely on the basis of judgment as to what this ratio should be for the most pleasing results.

In the design of the low frequency receiver one of the main objectives was to reduce to a minimum variations in sound transmission resulting from variations in the throat impedance of the horn. The high frequency horn can, however, be readily made of a size such that the throat impedance has relatively small variations within the transmitting region. On the other hand, while the diameter of the diaphragm of the low frequency unit is only a small fraction of the wave length, that of the high frequency unit will have to be several wave lengths at the higher frequencies in order to be capable of generating the desired amount of sound. Unless special provisions are made there will be a loss in efficiency because of differences in phase of the sound passing to the horn from various parts of the diaphragm. The high-frequency receiver was therefore constructed so that the sound generated by the diaphragm passes through a number of annular channels. There are a sufficient number of these channels to make the distance from any part of the diaphragm to the nearest channel a small fraction of a wave length. These channels are so proportioned that the sound waves coming through them have an amplitude and phase relation such that a substantially plane wave is formed at the throat of the horn.

The high frequency receiver unit was also similar in construction to the 555 receiver except that the channels connecting the diaphragm to the horn were redesigned so as to transmit frequencies up to 15,000 cycles. By the use of iron having high permeability at high flux density, and a compact ribbon-wound magnetizing coil considerably higher flux densities in the air gap were obtained resulting in an efficiency of over 50 per cent.

The microphones used for the transmission of music in acoustic perspective have been previously described

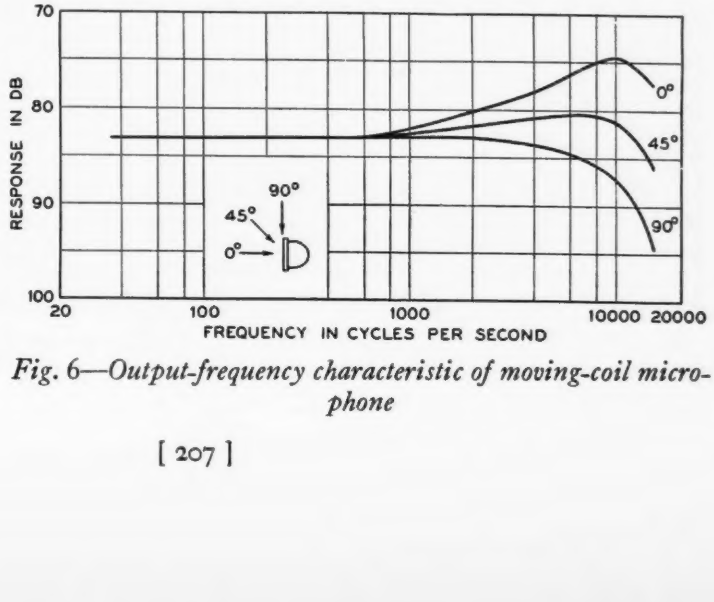
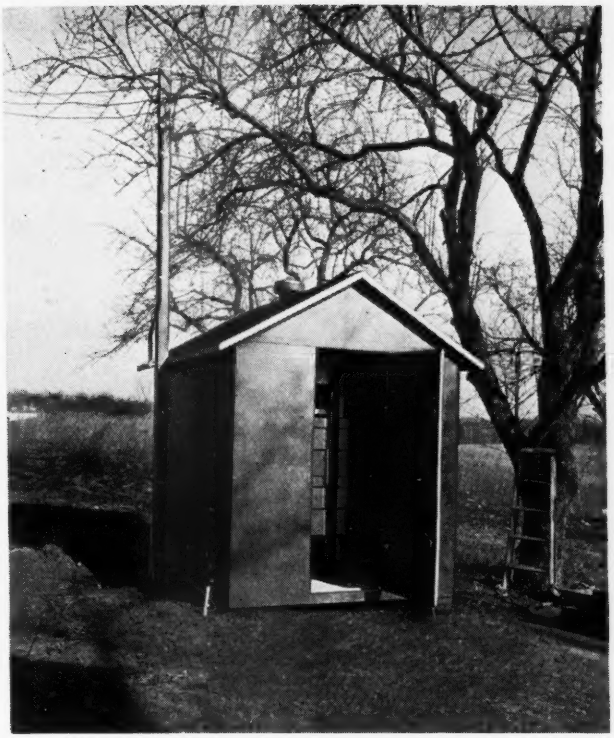


Fig. 6—Output-frequency characteristic of moving-coil microphone

in the Record.* They are of the moving coil pressure type. From their frequency-response characteristic as measured in an open sound field for several different angles of incidence of the sound wave on the diaphragm, it is seen that the response at the higher frequencies falls off as the angle of incidence is increased. This is, in general, not a desirable property, but with the orchestral instruments as here used, the sound observed as coming from each loud speaker is mainly that which is picked up directly in front of each microphone; sound waves incident at a large angle do not play a great part.

*Record, *May*, 1932, *p.* 314

At certain times the sound delivered by the orchestra is of very low intensity. It is therefore important that the microphones have a sensitivity as great as possible so that the resistance and amplifier noises may be kept down to a relatively low value. At a thousand cycles these microphones, without an amplifier, will deliver to a transmission line .05 micro-watts when actuated by a sound wave having an intensity of one micro-watt per square centimeter. This sensitivity is believed to be greater than that of microphones of other types having frequency response characteristics of comparable excellence.



Long Distance Transmission For Auditory Perspective

By R. E. CRANE
Toll Systems Development

THE main elements of the problem of transmitting electrical signals from one place to another are similar in kind for all applications. The objective is to reproduce at the distant end of the line a signal that is as nearly a duplicate of the original as can be economically justified for the purpose in hand. Perfection in attainment requires that the frequency and volume ranges of the original signal be faithfully transmitted, and in addition that no disturbances of any form be introduced by the transmitting system or allowed to enter from the outside. In general all the requirements placed on the transmission system become more severe as the signal becomes more complex.

Signals transmitted for the reproduction of music in auditory perspective place requirements on the transmitting system more severe than have ever been imposed before. In addition to the very wide frequency band, from 40 to 15,000 cycles, which is considerably wider than the 8,000 cycle program circuits—the highest grade circuits at the present time—there is the extremely wide volume range of 80 db. To obtain this wide volume range without using large amplifier tubes, it is necessary to allow the weaker passages of the music to reach extremely low levels. To make this

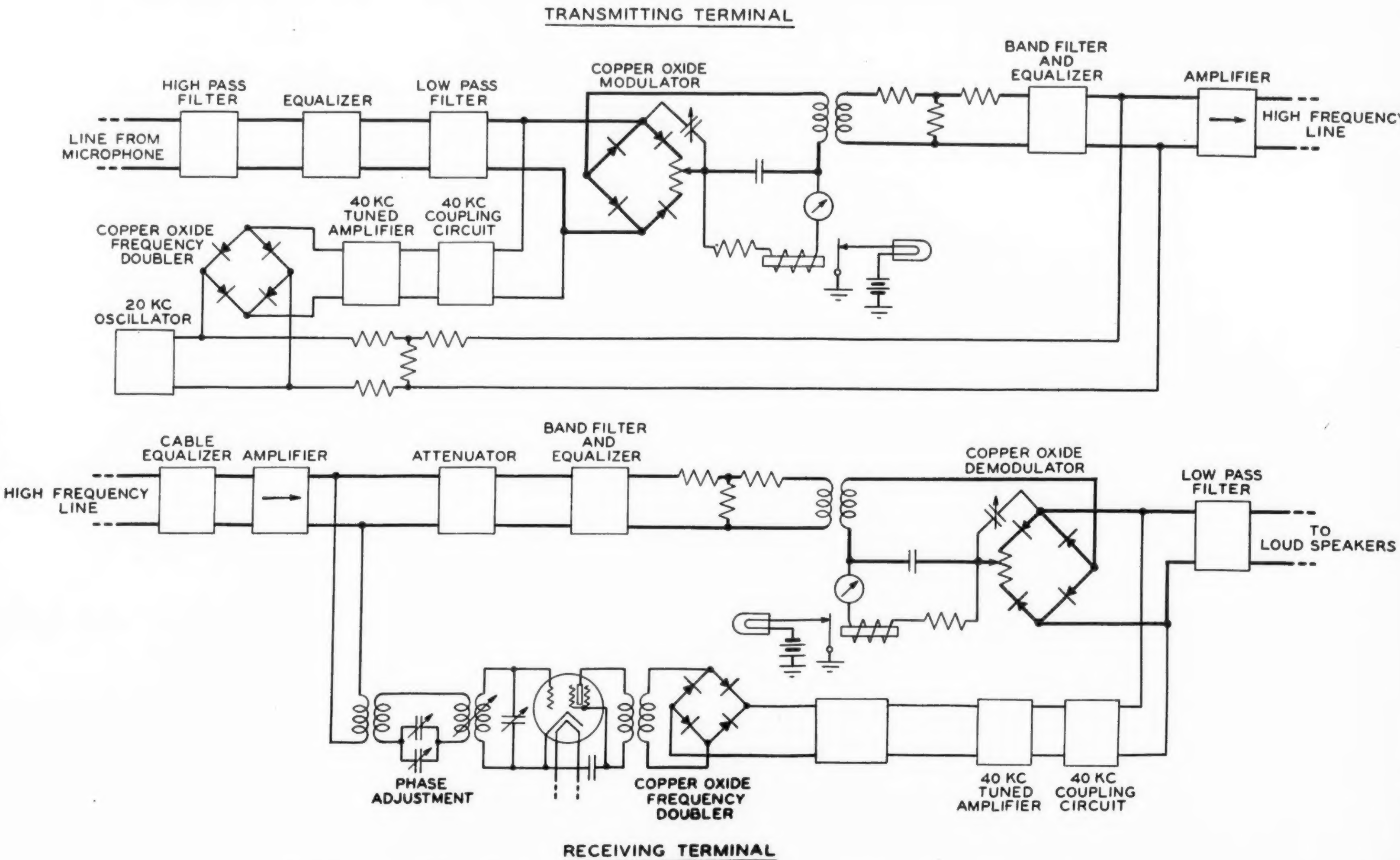


Fig. 1—*Simplified diagram of the circuit at the transmitting and receiving ends*

possible, in turn, all disturbing noises must be reduced to extremely low intensity to insure that they will always be lower than the lowest signal by a safe margin.

Still another difficulty arises from the necessity of providing three substantially identical circuits. The auditory perspective effect requires that there be no appreciable difference between the transmission characteristics of the three lines. Not only must the curves of loss against frequency show no variation greater than a small specified amount over the frequency range transmitted, but the shape of the curve for all three lines must be alike.

The system developed for this purpose makes use of carrier frequency transmission over toll cables. It was designed for a particular route between Philadelphia and Washington on which several 16-gauge non-loaded program pairs were available in the toll cables. A separate pair was used for each of the three circuits. Since the other pairs of the cable were operating at voice frequencies, the use of carrier for the auditory perspective circuits offered comparative freedom from cross-talk and noise within the cable.

The cable sheath, on the other hand, is very effective in shielding from outside interference at carrier frequencies.

Even under these favorable conditions it was found that at one office high frequency interference was being introduced into the cable longitudinally due to certain relay operations. To eliminate this disturbance special filtering arrangements were installed.

A carrier frequency of 40,000 cycles was employed, and the lower sideband—extending down to 25,000 cycles—was transmitted. The carrier itself was not transmitted, but exact synchronism between transmitting and receiving carriers was secured by deriving both from a 20,000 cycle frequency which was transmitted over the cable at low amplitude. This frequency is far enough below the signal band to be readily separated from it at the receiving terminal.

A single sideband is ordinarily obtained in a carrier system by inserting a band filter after the modulator to suppress one side-band and transmit the other. When the side-bands extend to within 40 cycles of the carrier frequency, as in this system, it is

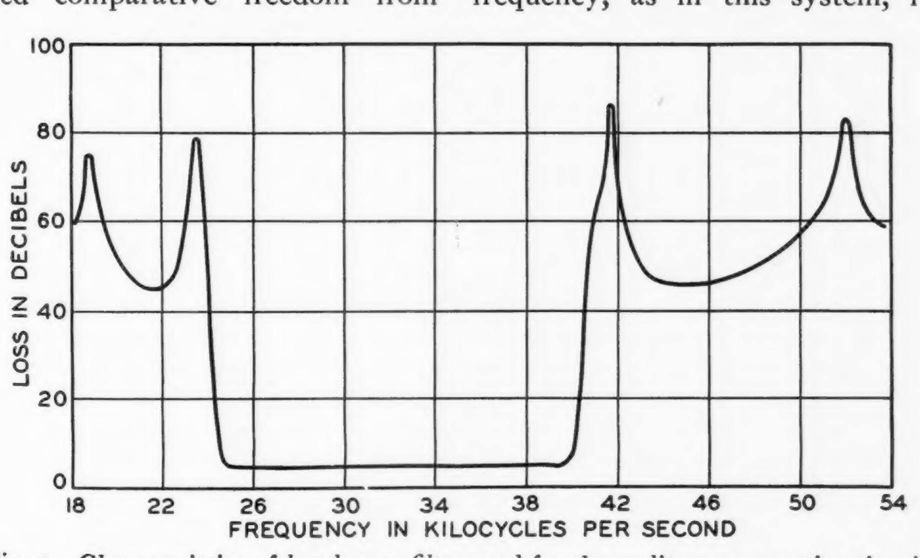


Fig. 2—Characteristics of band pass filter used for the auditory perspective circuits

extremely difficult to suppress completely these low frequencies of one sideband and yet transmit those of the other. An equivalent effect was obtained by using a filter of moderate selectivity in the neighborhood of the carrier frequency so that a portion of the unwanted sideband was transmitted along with the wanted sideband. At the receiving terminal the combination of this vestige of unwanted sideband and the partially suppressed wanted sideband gives the same result as a true single sideband by properly adjusting the phase of the receiving carrier.

The requirement is that the phase change for a given frequency interval above the carrier be equal and opposite to that of the same frequency interval below the carrier, and that the attenuation be arranged so that the arithmetic sum of the wanted and vestigial sideband for a given low frequency be substantially constant and equal to the amplitude of the frequency at midband. This method of course imposes special requirements on the phase and attenuation characteristics of the filters in the neighborhood of the carrier frequency.

For the modulating elements at both transmitting and receiving terminals, copper oxide rectifying discs were used. These elements can be made very simple and the stability of such circuits with respect to transmission and the ability to suppress the carrier frequency in balanced circuits is superior to the usual vacuum tube circuits.

Figure 1 shows schematically the arrangements of the carrier circuit at the transmitting and receiving ends. At the transmitting terminal the circuit from the microphone is first led through low and high pass filters to limit the band to the desired width, i.e., 40 to 15,000 cycles. The high pass filter was inserted to protect the systems from occasional high energy pulses of sub-audible frequencies. If these filters are omitted the carrier system will transmit down to and including zero frequency by virtue of the vestigial sideband technique. The rectifying discs are arranged to suppress the carrier frequency, the final degree of suppression being adjusted by means of the variable resistance and condenser shown.

The circuit includes a relay and meter both actuated by the rectified component of the carrier frequency. In this way a check on the magnitude of the carrier supply, and an alarm in case of its failure, is provided. The carrier frequency is derived as the second harmonic produced by an arrangement of copper oxide discs from a 20,000 cycle oscillator, part of the output of which is connected directly to the input of the transmitting amplifier.

Following the modulator is the band filter which, with the help of the receiving band filter, selects the sideband desired and provides the required characteristics in the vestigial sideband region. Figure 2 shows a transmission characteristic of this band filter. At the receiving terminal a modulation or demodulation process occurs

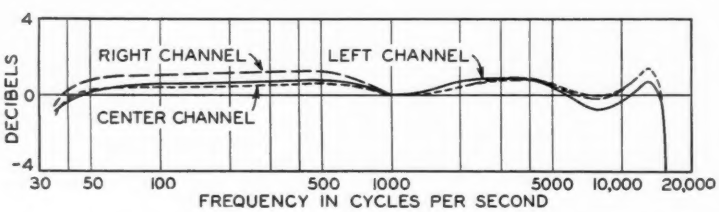


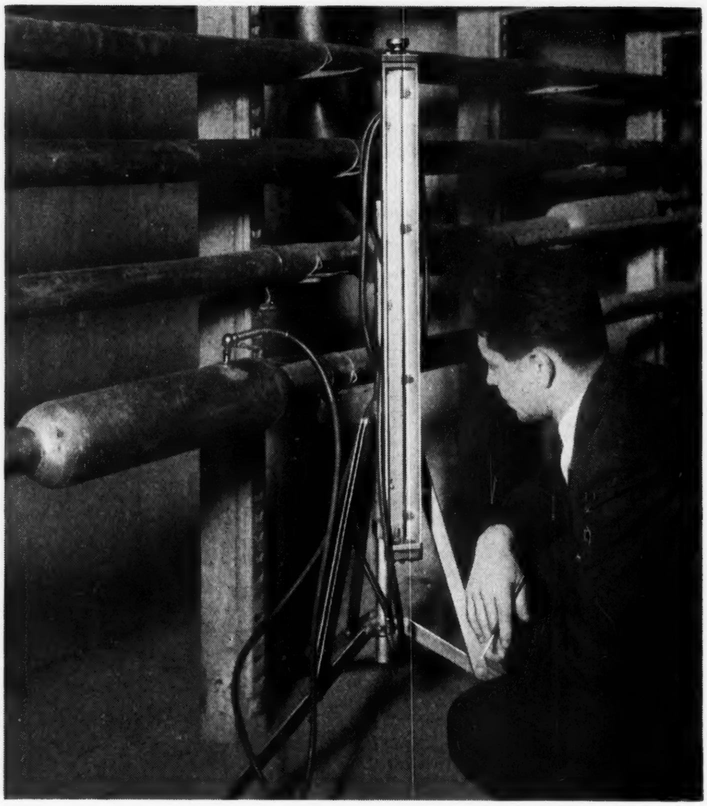
Fig. 3—Overall frequency transmission characteristics of the three circuits

similar to that at the transmitting terminal. The dc meter here also provides a check or pilot of the transmission over the long line circuit. The 20,000 cycle synchronizing current is selected at the receiving terminal, amplified, doubled, and supplied to the demodulator. The phase of the carrier is adjusted for proper vestigial sideband operation by means of a variable condenser included in the selecting circuit.

The attenuation of cable circuits at carrier frequencies is so great that repeater stations must be located at more frequent intervals than on voice-frequency cable routes. Between Philadelphia and Washington regular repeater stations are located at Elkton, Baltimore, and the two terminals. Three additional stations were established at Holly Oak, Abingdon, and Laurel. At Holly Oak, the amplifiers were located in an existing telephone office, but at the other two points small temporary structures were erected. That at Abingdon, which was of welded steel, is shown in the photograph at the head of this article.

The amplifiers used at intermediate repeater stations as well as at the terminals are of a new type characterized by remarkable stability with battery and tube variations and great freedom from non-linearity or modulation effects. Each amplifier is supplemented at its input by an equalizer having a loss characteristic approximately complementary to that of the preceding cable section. Variations in loss of the cable sections, all of which are slow and due to temperature changes, are compensated by adjustments of the variable equalizer arrangements provided.

The complete carrier system was operated over a period of several weeks between Philadelphia and Washington, during which demonstrations were held. The overall frequency transmission characteristics of the three channels are shown in Figure 3. These curves include the complete high frequency line circuit with its 150 miles of cable, repeaters, equalizers, etc. It will be noted that the three are substantially equal and are flat in transmission performance between the desired frequency limits to within ± 1 db. Noise introduced by the line and apparatus was inaudible in the auditorium in Washington even during the weakest music passages. With this system, the reproduction of symphonic music in Washington was identical as far as could be determined with a reproduction of the same program in an auditorium located in the same building in which the orchestra was playing.



Pounds of Prevention—Gas-Filled Cables

By H. BAILLARD
Outside Plant Development

THE practical application of the old adage "An ounce of prevention is worth a pound of cure" is well illustrated in present day telephone plant practices. Comprehensive programs of preventive maintenance afford a control of much of the plant apparatus which is closely associated with the preservation of uninterrupted service. The accompanying routine tests and inspections are directed at the location and correction of plant trouble in its incipient stage, and have proved most valuable in the elimination of service failures. Prominent examples of this work may be seen in the automatic routine tests of dial central office equipment and the frequent voltmeter tests of subscriber lines. In the outside plant an increasing amount of toll cable is being equipped with a protective alarm system which both warns of accidents to the lead sheath and forestalls the circuit trouble which might otherwise follow from the entrance of moisture. At the present time this system is used on a high proportion of the underground and to a somewhat lesser extent on the aerial toll cable in the Bell System.

Because of the protection afforded by the lead sheath and the decreased susceptibility to storm and other

forms of damage, cables provide the most dependable type of telephone circuit. However, it is unfortunately true that accidental punctures of the lead sheath are far from uncommon and they immediately present a potential if not always actual path for the entrance of moist air or water. Since moisture will quickly destroy the effectiveness of the paper insulation surrounding each conductor, it is important that sheath openings be located and repaired as quickly as possible.

Prior to the introduction of the protective alarm system, direct inspection was the only means of detecting the presence of sheath breaks, besides electrical measurements made after actual conductor trouble developed. As the cables are usually in underground conduit or in aerial runs, the accessibility for inspection leaves much to be desired. In the first case, since the only access is at manholes, thorough visual examination is out of the question, and in the second case, a tedious aerial cable "ride" is involved. Even in favorable circumstances the most painstaking examination will sometimes fail to reveal a minute opening in the lead sheath.

The basis of the newly developed protective system is the continuous maintenance of the cable under gas pressure. When a sheath failure occurs, the outrush of gas will naturally drive back incoming air or water. The drop of pressure inside the cable following the loss of gas through a leak operates an alarm mechanism connected by wire to the repeater station. This system, providing the double service and involving the use of specially designed apparatus, is installed as a permanent part of the cable plant.

A development associated with the pressure maintenance of toll cable is a testing procedure for detecting defects in the solder work or other parts of newly installed cable, either exchange or toll. This check-up is applied independently of the ultimate maintenance of the installation and is widely used on non-pressure maintained systems.

It is interesting to note at this point some of the causes of lead sheath puncture. With aerial cable, a frequent cause is gun-shot, usually but not invariably resulting from carelessness. Aerial cable may also be damaged by flying stone fragments from incompletely shielded blasting operations; by storms and their accompaniment of wind-driven branches and falling trees, and strangely enough, by the boring of certain insects and by the gnawing of squirrels. Cables in underground ducts may suffer from such things as digging or blasting during highway or pipe-line construction and from corrosion. This corrosion may arise from a multitude of causes varying from immersion in salt water from a neigh-

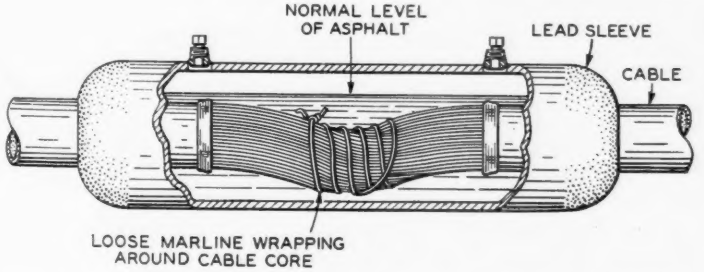


Fig. 1—In constructing the gas-tight plugs, the core is impregnated with a specially selected wax and the surrounding lead sleeve is nearly filled with asphalt introduced as a hot liquid. At normal atmospheric temperatures the asphalt is semi-plastic and forms a gas-tight seal.

boring marsh to the electrolytic action of stray trolley line currents.

Cables at river crossings are exposed to perhaps the most violent mishaps of all, where a ship's dragging anchor can gouge very sizable holes. There are cases on record, however, in which the outflow of gas following such severe injury prevented the entrance of water for considerable periods and so limited the damage that repairs were relatively easy when the cables were hauled up from the river bottom.

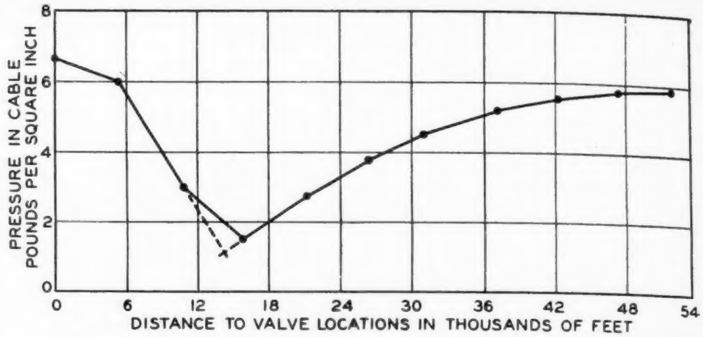


Fig. 2—The exact location of a leak is determined by means of the pressure gradient along the affected part of the cable. Points on the graph indicate positions of the valves where pressure readings were taken

Finally, the highly developed technique of solder wiping at cable splices may, like any other manual operation, show an occasional imperfection. The pinhole or pore thus left is a potential entrance for moisture.

To understand the functioning of the protective alarm system something must be noted of the structure of the cable itself. Although a cross-section cut shows an area apparently solidly filled with copper wires and their paper insulation, about half of the cable volume is actually air space. In a cable maintained with the protective system this volume forms the gas reservoir and is charged with dry nitrogen gas to a pressure of about 9 pounds per square inch. Nitrogen is used because it is relatively inexpensive and can be obtained in a very dry condition.

By means of gas dams, called "pressure testing plugs", a cable is divided into individually gas-tight sections. These usually are between seven and eleven miles in length. Such sectionalizing facilitates charging, testing, and repair operations, and provides a sufficient reservoir of gas. The attached illustrative drawing shows the essentials of a plug.

Each gas-tight cable section is equipped with from three to five of the alarm mechanisms, with each of which is associated a terminal suitable for making connections to the alarm pair and an associated cableman's talking pair in the cable. These mechanisms, called pressure contactors, operate on the fundamental principle of the ordinary pressure gauge. In the accompanying photograph, a contactor is shown with its gas-tight cover removed. In use, a contactor is connected to a terminal which in turn is connected to the cable, so that the pressure of the gas surrounding the "Bourdon" tube is always that of the cable itself. The degree of curvature of this sealed tube is controlled by the cable pressure, the movement of the tube end being amplified by a linkage to actuate the contact springs.

Through the periods during which the gas pressure in the cable is above the value at which the alarm operates the contact points are held apart. However if the sheath be punctured, the resulting drop of gas pressure permits the contacts to close, and the

alarm at the repeater station operates. All of the contactors are connected in parallel across a single cable pair. Resistance measurements made on this pair from the repeater station by means of a Wheatstone bridge are

Fig. 3—Pressure contactor with gas-tight cover removed

used to "locate" an operated contactor, and thus the approximate position of the leak.

Cable sheath troubles are accompanied by a relatively slow escape of gas, therefore the internal pressure drop is correspondingly slow. This is mainly due to the impedance offered to gas flow by the closely packed paper and copper forming the cable core. This impedance combined with the large actual volume of gas contained in a gas-tight section, produces a condition of graduated pressures starting with a low point at the break and rising in both directions to a maximum at the ends of the section. It is interesting to note that the impedance mentioned above so restrains gas flow that a completely severed end loses gas at a rate not appreciably greater than a relatively small hole. The drop of pressure resulting from a leak at the mid-point of a section may not be evident at a plug for some time.

After a leak has been detected by operation of a contactor and the contactor has been located by Wheatstone bridge measurements, pressure measurements are made in the vicinity of the operated contactor by a cableman. To facilitate this, valves somewhat similar to automobile tire valves are permanently installed along the cable at intervals of about one-half mile. If the leak is large a gauge of the Bourdon tube type is used for the pressure measurements, while in the case of a small leak, a sensitive mercury manometer is employed as shown in the photograph at the head of this article. The pressures thus measured are plotted against distance to form a graphic picture of the pressure gradients, one on each side of and sloping towards the low pressure point. By extending these gradients to their intersection the location of the leak is very closely approximated. The attached drawing shows a typical pressure gradient chart.

In general, locations made by the Bourdon tube type gauge or the manometer are accurate enough to indicate that the leak is in a particular splicing section of cable. In underground plant, where the cable is available for inspection only at manholes, it is desirable to make check tests of the location before pulling out and replacing the suspected manhole section or digging up the section for repair. For this purpose a device known as a gas flow indicator is used. The indicator

consists essentially of a valve containing an ammonia chamber placed between two sight tubes that have hose connections at their opposite ends. Ammonia is placed in the ammonia chamber, and blotting paper strips saturated with phenolphthalein solution are placed in the sight tubes. The hoses are then connected to valves installed several feet apart on the cable. Gas flows through the indicator in the same direction that it is flowing through the cable, thus carrying ammonia vapor from the ammonia chamber through one of the sight tubes and causing the paper strips in that tube to change color.

In the cases of leaks in aerial cable or at accessible places in underground cable, final determination of the exact point of leakage is usually made by painting the suspected portion of the cable with soap solution and watching for bubbles. A specially compounded soap solution yields bubbles having superior film strength and durability. This method of locating leaks is also applied to the preliminary pressure testing of new installations, as previously described. In testing an individual cable sleeve, for instance, gas is admitted to the sleeve until a localized pressure is built up, and the wiped joints and soldered seam are then painted with soap solution. Any defects in the solder work are thus detected.

The sheath break having been repaired, the cable is recharged to its maintenance pressure and the protective alarm system once more stands on guard.

Western Electric Noiseless Recording

By W. A. MacNAIR
Special Products

IMPROVEMENTS in recording sound on film depend upon the development of improved apparatus and the refinement of technic along many lines. Extension of the frequency range is desirable to cover more adequately the audible frequency spectrum. Increased volume range would be of benefit in more nearly accommodating the wide range of sound levels often produced by a symphony orchestra. Frequency modulation, which appears whenever the speed of film during the recording or reproducing process is not constant, must be kept to a low value. Non-linear distortion must be minimized.

In this series of statements, problems have been segregated which are really not entirely independent of one another. For instance, as the frequency range is extended, non-linear distortion must be more carefully watched, less frequency modulation is tolerable, and the problem of increased volume range becomes more urgent. The segregation is convenient, however, for purposes of study and discussion.

The extension of the normal volume range* in the records of sound on film has had wide public notice in recent years. Since the special apparatus which accomplishes this is used in the recording operation, without necessitating any modification of reproducing equipment, the commercial process for making records of this improved type with Western Electric sound-on-film recording systems is known as "Noiseless Recording".

**In this use, the term "volume range" may be defined as the difference in level between the reproduced record of maximum modulated 1000 cps signal and the noise of unmodulated sound track.*

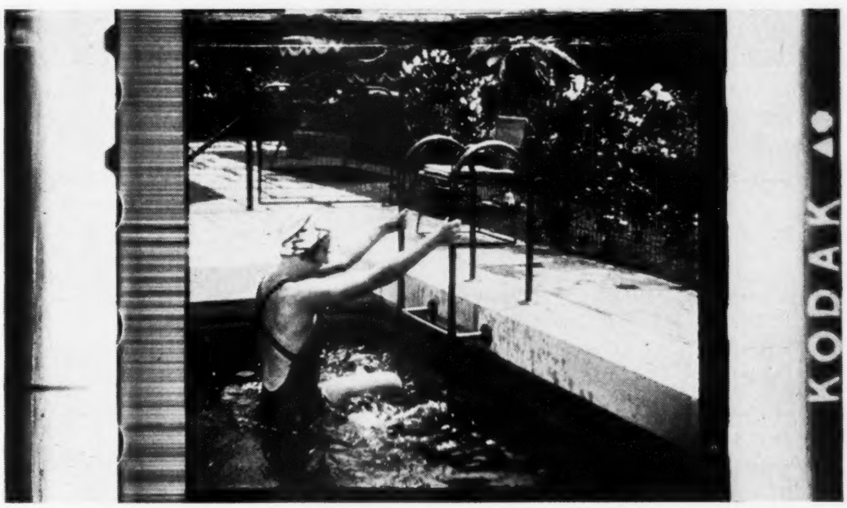


Fig. 1—Western Electric Noiseless Recording makes it possible to reproduce the sounds of dripping water, otherwise masked by background noise

In the photographic technic of sound-on-film recording—the exposure and development of the negative, and the printing and development of the positive—it is the aim that the variations in the quantity of light falling on the photoelectric cell in the reproducer at various instants shall be exactly proportional to the variations in the quantity of light exposing the film at the corresponding times during recording. If at some instant the sound current has opened up the light valve to twice its average dimensions, then the point on the sound print corresponding to that instant allows twice the average quantity of light to fall on the photoelectric cell in the reproducer.* Thus, regarded in the simplest terms, the film record is a means of storing away, in reproducible form, the wave shape of the variations of the quantity of light passed by the light valve.

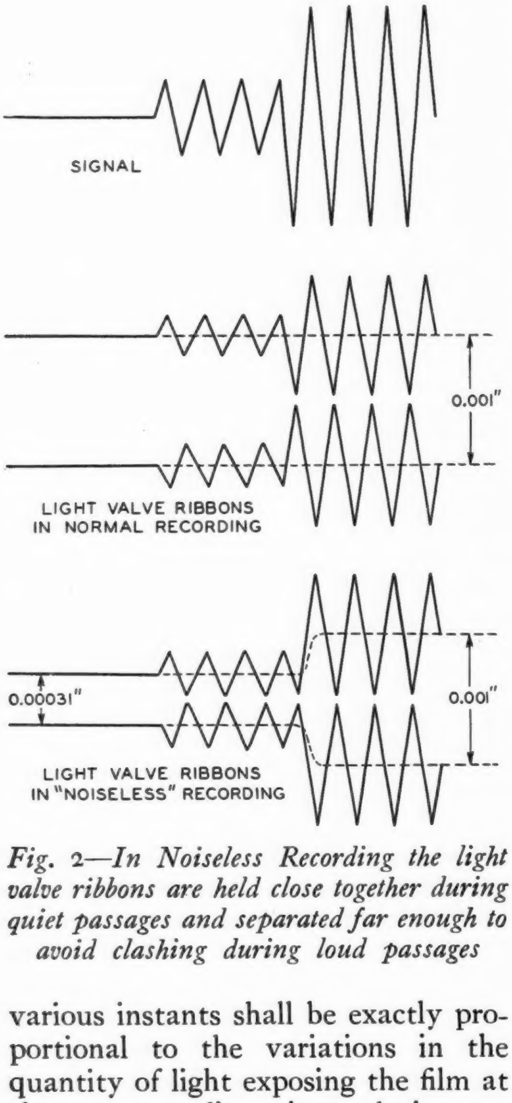


Fig. 2—In Noiseless Recording the light valve ribbons are held close together during quiet passages and separated far enough to avoid clashing during loud passages

In the light valve**, which is the heart of the recording system, a loop of duralumin tape is mounted under tension so that, in the center portion of the loop, the two ribbons lie in the same plane, with the inner edges parallel and separated one mil (one thousandth inch). This center portion is located in a strong magnetic field, and the sound signal from the recording amplifiers, passing through the loop, causes the ribbons to move nearer to and farther away from each other in accordance with the current through them. During recording, the sound currents are supplied to the valve at a level such that the maximum peaks fully modulate the valve: that is, so that the peak amplitude of the valve strings, or the double amplitude of each string, is just one mil.

Since the wave shape is one of variations about an average, it makes little difference what the value of the average is. Variations of light about an average value are analogous to a unidirectional pulsing current, and indeed this is what these variations produce in the output of the photoelectric cell in the reproducer. Such a current can be considered as the sum of alternating and direct current components, and the direct current component, for which the average value of light is responsible, is removed by transformers before reaching the loudspeakers.

**In this case the greater quantity of light passed by the light valve produces a DARKER portion of the negative which when printed yields a LIGHTER portion of the positive.*

**RECORD, *November*, 1928, *p.* 95, *and August*, 1932, *p.* 412.

In assuring faithful duplication of the wave shape, however, it is important that the average amount of light be at least as great as the largest variation required. There is nothing darker than complete darkness; or what amounts to the same thing, when the light valve is completely closed, it cannot be closed any further.

It is the passage of the large average amount of light required to accommodate these maximum variations that has caused the background noise now reduced by "Noiseless Recording". Curiously enough the magnitude of the background noise which is present in the reproduced signal from a sound-on-film record has been found to be intimately connected with the amount of light transmitted by the sound track. It is an empirical fact that, other things remaining constant, the noise level reproduced is lowered six decibels if the light transmission is cut in half.* This relationship holds throughout the range of transmissions at present employed in commercial "noiseless" sound-on-film records. Since the reproduced noise level depends upon the light transmission of the print, and this in turn depends upon the spacing of the light-valve ribbons, the noise from the print can be controlled by the spacing of the ribbons. This is exactly what is done in the "Noiseless Recording" process.

It is of considerable importance to discover the fundamental causes of film noise and the factors whose control will minimize it.

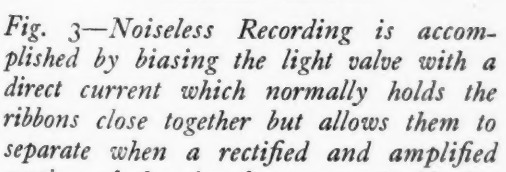


Fig. 3—Noiseless Recording is accomplished by biasing the light valve with a direct current which normally holds the ribbons close together but allows them to separate when a rectified and amplified portion of the signal opposes the biasing current

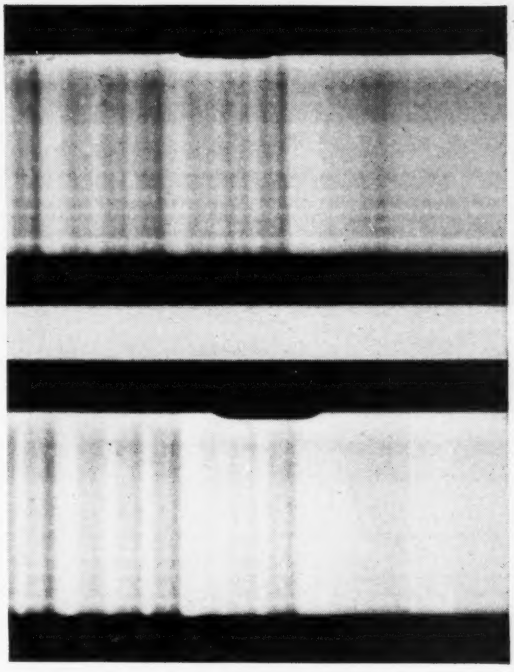


Fig. 4—A positive print from a negative made by Noiseless Recording (above) is darker during the quiet passages than a print made in the usual way (below)

In recording both speech and music, it is well known that during a great portion of the time the excursions of the ribbons are far less than the extreme value. At times when the excursions are small, there is no necessity for the average spacing of the ribbons to be as great as one mil; the spacing need be only sufficient to avoid clashing of the ribbons. It is this allowable latitude in the adjustment of spacing which permits the application of the principle of the biased light valve to produce "Noiseless Recording".

In this process the usual one-mil light valve is closed down to a spacing of about one-third mil by means of a

direct current in the ribbons. Reducing the spacing of the strings to one-third reduces the light transmission of the print to one-third and lowers the reproduced noise level about ten decibels. All of the low-level portions of the sound record are made with this reduced spacing of the ribbons.

When a portion of the sound to be recorded is of a higher level than this spacing will accommodate, the spacing is momentarily increased to prevent clash. This operation is accomplished automatically by amplifying a portion of the signal being recorded, rectifying the amplified current, and supplying the rectified signal to the light-valve ribbons thru a suitable filter. During sustained portions of high level signals the valve might stay fully open to one mil for a considerable time, after which the spacing would decrease automatically to one-third mil as the sound level decreased.

The print thus produced is such that during the low level and silent portions of the record, the reproduced noise level is ten decibels lower than would have been the case if the valve spacing had not been reduced. During the high level portions of signal the noise is just as it would have been, since the valve is opened to one mil during such passages. Background noise, however, is not as noticeable during the loud portions of the reproduced sound as during the low level and silent portions. By closing the valve during these latter periods, the recording process reduces the background noise precisely where it would be most objectionable.

The use of "Noiseless Recording" has considerably enhanced the dramatic value of recent sound pictures. Such sounds as faint foot steps or a key clicking into a lock are used to far greater advantage on film records with the lower background noise which "Noiseless Recording" makes possible.

Contributors to This Issue

E. H. Bedell received the B. S. degree from Drury College in 1924 and then spent a year doing graduate work at the University of Missouri. The following year he joined the Laboratories as a member of the Acoustical Research department. Here his work has had to do mainly with studies of sound absorption and transmission and allied subjects in the field of architectural acoustics.

W. B. Snow is a graduate of Stanford University with A. B. and E. E. degrees. He became associated with the Laboratories in 1923 and left nearly a year later for further studies at Stanford. In 1925 he returned to the Laboratories as a member of the Acoustical Research group. He has been engaged chiefly on investigations of distortion in speech and music.

R. E. Crane graduated from the Harvard Engineering School in 1923 and immediately joined the Engineering Department of the Western Electric Company, now the Laboratories. He first engaged in the development of the terminals for the type "C" open-wire carrier telephone system. He also assisted in the field tests of several models of that system and later in the supervision of testing the early commercial product. His work has been continuously concerned with carrier systems, the last few years in charge of a group developing terminals for cable carrier and special purpose carrier systems.

H. Baillard received an A. B. from Columbia University in 1923, and the degree of Chemical Engineer, from Columbia's School of Engineering, in 1925. The following year he joined the Technical Staff of Bell Telephone Laboratories where, with the Chemical Department, he engaged in studies of electroplating and corrosion, enameled wire, and miscellaneous organic materials. In 1928 he transferred to the Outside Plant Department. Here in addition to certain staff work he has worked on paint problems, and more recently on cable joining and maintenance.

Receiving the B. S. degree from Colgate in 1920, W. A. MacNair spent a year as assistant physicist with the Bureau of Standards, interesting himself with aeronautical instruments, and a year as

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instructor in mathematics and physics at the Michigan College of Mines. Two years of further study at Johns Hopkins University brought him the M.S. and Ph. D. degrees in physics and a National Research Council Fellowship. Returning to the staff of the Bureau of Standards in 1927, he worked for a year on atomic structure as associate physicist, then went to the research division of the Victor Talking Machine Company. In 1929 he joined Bell Telephone Laboratories' Sound Picture Laboratory where he now has charge of a group investigating acoustical problems in sound recording and reproduction.

A. L. Thuras received a B. S. and an E. E. degree in Electrical Engineering from the Univesity of Minnesota in 1912 and 1913 and then joined the staff of the Bureau of Standards in Washington. He served as Scientific Observer on an expedition to Newfoundland and from 1916 to 1920 was Oceanographer for the Coast Guard. This work consisted in the development and use of continuous recording instruments for measuring the temperature, salinity, and density of sea water. Records were made chiefly in the vicinity of the Great Bank of Newfoundland, and were used to locate and study the water movement of the Labrador Current and the Gulf Stream. In 1920 he joined the technical staff of the Laboratories where he has been engaged in the study and development of electro-acoustical instruments.